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(54) Title: ADDRESS DEFINITION FOR IP TELEPHONY SERVICES

(57) Abstract: A method of providing IP telephony services maps a telephony signaling protocol called party number nature of address indicator (ISUP IAM) to an Internet signaling protocol nature of address indicator (SIP INVITE) and a telephony signaling protocol called party number numbering plan indicator (CIRCUIT) to an Internet signaling protocol numbering plan indicator (RTP).

ADDRESS DEFINITION FOR IP TELEPHONY SERVICES

BACKGROUND

The present invention relates generally to the field of Internet telephony, and more particularly to a method of and system for providing address definition for public switched network terminated calls that use Internet telephony services.

DESCRIPTION OF THE PRIOR ART

The existing public switched telephone network (PSTN) environment provides a rich set of telephony services. In order to work properly, the services depend on address information that is carried as part of the call setup process. This address information includes the called party number and two qualifiers to the called party number. These qualifiers are the nature of address (NOA) and the numbering plan indicator (NPI).

The called party number and the qualifiers are used in the PSTN to determine how to handle calls. The nature of address can be subscriber, national, or international. The numbering plan indicator can be public (E.164) or private. A dialed number address with NOA and NPI values indicating the called party number is an international public number will be routed differently than the same number that has NOA and NPI values indicating that it is in a private numbering plan.

Efforts are currently underway to extend the services to devices connected to Internet protocol (IP) networks. One protocol that has been developed to handle the called related signaling in the IP based networks is the session initiation protocol (SIP). The addressing mechanism built into the SIP specification supports only global and local numbers. The SIP specification does not support the NOA and NPI qualifiers used in the PSTN. As a result, is not possible to support all of the existing PSTN services when one leg of a call must pass through an IP network or terminate to the device on an IP network.

SUMMARY

The present invention provides a method of and system for providing IP telephony services by mapping a telephony signaling protocol called party number nature of address

indicator to an Internet signaling protocol nature of address indicator and mapping a telephony signaling protocol called party number numbering plan indicator to an Internet signaling protocol numbering plan indicator. The present invention is preferably implemented by extending the session initiation protocol (SIP) invite request called party URL to include a nature of address parameter and a numbering plan parameter. The system of the present invention provides a gateway that performs the mapping.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram of an Internet telephony system.

Fig. 2 is a block diagram of an Internet telephony gateway.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring now to the drawings, and first to Fig. 1, a telecommunications system is designated generally by the numeral 11. System 11 includes a plurality of networks that are interconnected through an IP network 13, which in the preferred embodiment is the Internet.

The networks of system 11 include, in addition to IP network 13, a private intranet 15, and Internet service provider network 17, a public switched telephone network 19, and a private telephone network designated generally by the numeral 21. All of the networks in system 11 are adapted to provide IP telephony services.

In IP telephony, at least a portion of a call is transported over an IP network. Call setup and tear down are supported by an IP signaling protocol such as session initiation protocol (SIP) or H.323. The preferred embodiment of the present invention is implemented using the SIP signaling protocol. Media transport is supported by an IP protocol such as real-time transport protocol (RTP).

In the environment of private intranet 15, IP telephony enabled devices, such as IP enabled personal computers 23 and IP phones 25, are connected to private intranet 15 through local area network (LAN) connections. Personal computers 23 and IP phones 25 are SIP clients or user agents. They are provided with software that enables them to initiate and participate in IP telephony sessions. Similarly, in the environment of Internet service provider network 17, IP telephony devices, such as personal computers 27 and IP phones 29, are operably connected to Internet service provider network 17, typically through dial-up modem connections. Again,

personal computers 27 and IP phones 29 are SIP clients or user agents, which allows them to initiate and participate in IP telephony sessions.

In the environments of private telephone network 21 and public switched telephone network 19, devices such as telephones 31 and 33 do not have the native capacity to participate in IP telephony sessions. Networks such as private network 21 and PSTN 19 use traditional telephony signaling protocols such as SS7, ISDN, or channel associated signaling (CAS) for call setup and tear down. Networks such as PSTN 19 and private network 21 use traditional time division multiplexed circuits for media transport. In order for users and networks such as PSTN 19 and private network 21 to participate in IP telephony sessions, a gateway to an IP network must be provided. In the case up PSTN 19, a PSTN/IP gateway 35 is provided. In the case of private network 21, an enterprise gateway 37 is provided.

Referring to the Fig. 2, an IP telephony gateway is shown at 39. Gateway 39 includes a signaling gateway 41 and the media gateway 43. Generally, signaling gateway 41 provides bidirectional protocol translation between a telephony signaling protocol, such as SS7, and an IP telephony signaling protocol, such as SIP. As shown Fig. 2, signaling gateway 41 is adapted to provide bidirectional translation between, for example, ISDN user part (ISUP) initial address messages (IAM) and SIP invite requests. Media gateway 43 is adapted to provide bidirectional protocol translation between traditional time division multiplexed telephony circuits and real-time transport protocol packets.

Traditional telephony has developed a rich set of services with a rather limited addressing scheme. The traditional telephone user is able only to enter digits. Public switched telephone network switches and intelligent network elements are able to interpret the entered digits with the aid of a nature of address (NOA) indicator and a numbering plan indicator (NPI). The NOA and NPI enable the switches and intelligent network elements to correctly interpret the called party number.

As an example of public switched telephone network addressing, consider the digits 17035551234, NOA equal international, NPI equal public. This example indicates an international call to country code 1, area code 703, subscriber 555-1234. The same digits 17035551234 with NOA equal subscriber and NPI equal private indicates a special call subscriber 17035551234. A public switched telephone network switch can interpret the first example and route the call appropriately. Since the second example uses a private numbering plan, the switch

must consult a database, such as a service control point (SCP) for routing instructions.

Session initiation protocol does addressing by mapping telephone numbers to URLs. SIP URLs are of the form Atelephone_number@gateway. For example, telephone number 17035551234 maps to SIP URL SIP:17035551234@xyz.com. Currently, the SIP standard defines only two types of telephony URLs -- global and local. The global URL is of the form country code plus national number. The local URL is anything that is not global. Currently, signaling gateways ignore NOAs and NPIs, and map only the dialed digits portion of an SS7 called party number to a SIP to URL. Thus, SIP can interpret the number 17035551234 only as an international call.

According to the present invention, SIP is extended to include an NOA parameter and an NPI parameter in telephony URLs. Thus gateways according to the present invention map the entire called party number, including the NOA and NPI, to a SIP URL. For example, an SS7 IAM message with a called party number 17035551234, NOA = subscriber, NPI = private maps to a SIP invite request of the form:

INVITE:17035551234@xyz.com,NOA=subscriber,NPI=private

FROM:7774321@abc.com

TO:17035551234@xyz.com,NOA=subscriber,NPI=private.

The present invention thus enables the public switched telephone network to properly interpret telephone numbers that have been transported via IP telephony signaling and provide appropriate services. The present invention allows IP telephony to take advantage of the existing set of service provided by the PSTN.

WHAT IS CLAIMED IS:

1. 1. A method of providing IP telephony services, which comprises the steps of:
2. mapping a telephony signaling protocol called party number nature of address indicator to
3. an Internet signaling protocol nature of address indicator; and
4. mapping a telephony signaling protocol called party number numbering plan indicator to
5. an Internet signaling protocol numbering plan indicator.

1. 2. The method as claimed in claim 1, wherein said Internet signaling protocol is session
2. initiation protocol.

1. 3. The method as claimed in claim 2, wherein said Internet protocol nature of address
2. indicator and said Internet protocol number plan indicator are contained in an invite request.

1. 4. The method as claimed in claim 1, wherein said telephony signaling protocol is SS7.

1. 5. The method as claimed in claim 4, wherein said telephony signaling protocol is ISUP.

1. 6. The method as claimed in claim 1, wherein said telephony signaling protocol is ISDN.

1. 7. The method as claimed in claim 1, wherein said telephony signaling protocol is CAS.

1. 8. The method as claimed in claim 5, wherein said telephony protocol nature of address
2. indicator and said telephony protocol number plan indicator are contained in an initial address
3. message.

1   9.   A method of providing telephony services, which comprises the steps of:

2   receiving a call setup message in a first signaling protocol at a gateway between a first

3   network and a second network, said call setup message including a called party number, said

4   called party number including a numbering plan indicator and a nature of address indicator; and

5   generating a call setup message in a second signaling protocol at said gateway between

6   said first network and said second network, said call setup message in said second signaling

7   protocol including said called party number, said called party number including said numbering

8   plan indicator and said nature of address indicator.

1   10.   The method as claimed in claim 7, wherein said first network is a public switched

2   telephone network and said second network is an IP network.

1   11.   The method as claimed in claim 8, wherein said first signaling protocol is SS7 and said

2   second signaling protocol is session initiation protocol.

1   12.   The method as claimed in claim 9, wherein call setup message in said first protocol is an

2   initial address message and said call setup message in said second protocol is an invite request.

1   13.   The method as claimed in claim 7, wherein said first network is an IP network and said

2   second network is a public switched telephone network.

1   14.   The method as claimed in claim 11, wherein said first signaling protocol is session

2   initiation protocol and said second signaling protocol is SS7.

1   15.   The method as claimed in claim 11, wherein said first signaling protocol is session

2   initiation protocol and said second signaling protocol is ISDN.

1   16.   The method as claimed in claim 11, wherein said first signaling protocol is session

2   initiation protocol and said second signaling protocol is CAS.

1   17.   The method as claimed in claim 12, wherein call setup message in said first protocol is an

2   invite request and said call setup message in said second protocol is an initial address message.

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1   18.   An IP telephony gateway, which comprises:

2   means for mapping a telephony signaling protocol called party number nature of address

3   indicator to an Internet signaling protocol nature of address indicator; and

4   means for mapping a telephony signaling protocol called party number numbering plan

5   indicator to an Internet signaling protocol numbering plan indicator.

1   19.   The IP telephony gateway as claimed in claim 14, wherein said Internet signaling protocol

2   is session initiation protocol.

1   20.   The IP telephony gateway as claimed in claim 15, wherein said telephony signaling

2   protocol is SS7.

1   21.   The IP telephony gateway as claimed in claim 15, wherein said telephony signaling

2   protocol is ISDN.

1   22.   The IP telephony gateway as claimed in claim 15, wherein said telephony signaling

2   protocol is CAS.

1   23.   The IP telephony gateway as claimed in claim 16, wherein said telephony signaling

2   protocol is ISUP.

1   24.   The IP telephony gateway as claimed in claim 14, comprising:

2   means for mapping an Internet signaling protocol called party number nature of address

3   indicator to a telephony signaling protocol nature of address indicator; and

4   means for mapping an Internet signaling protocol called party number numbering plan

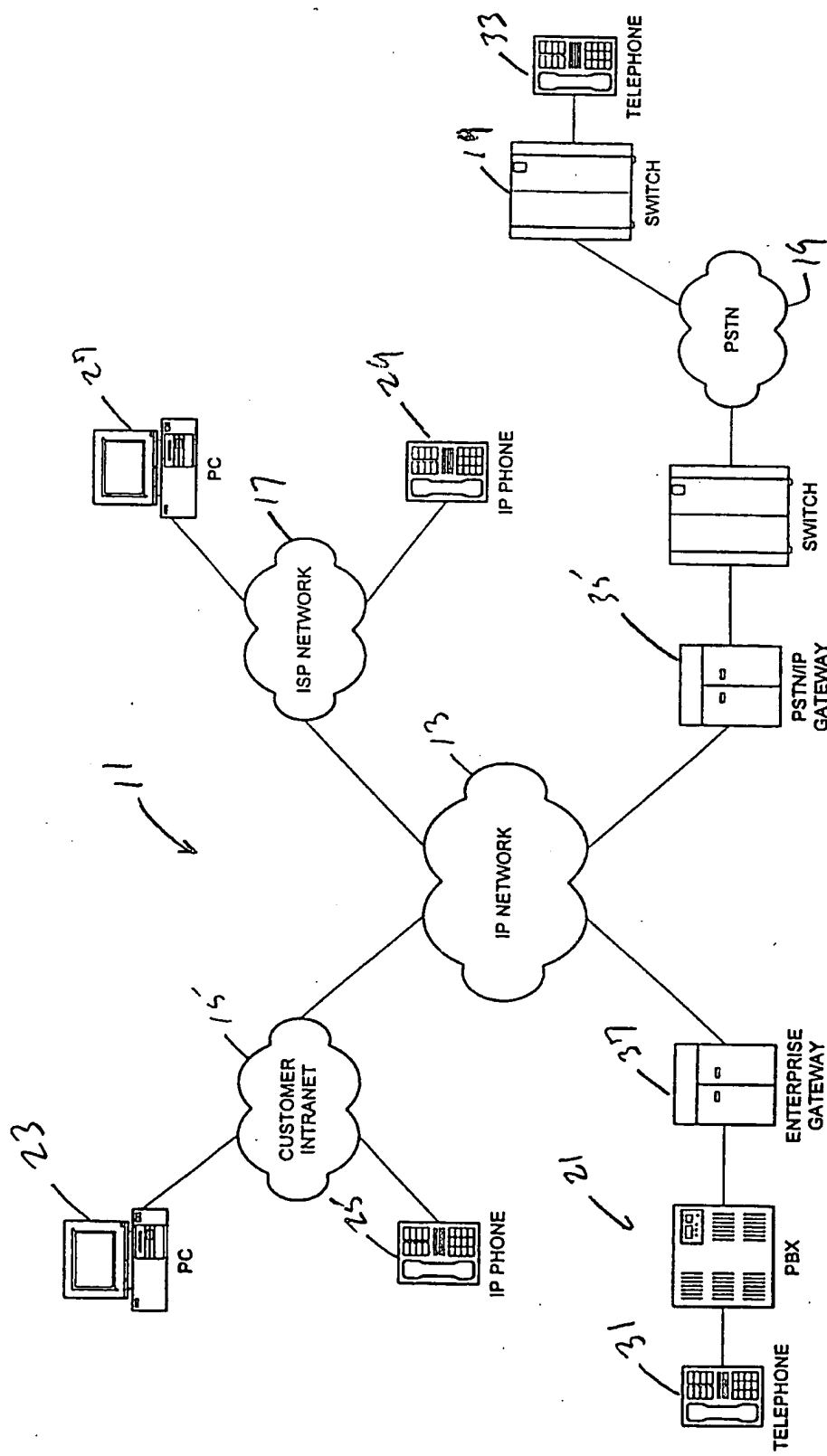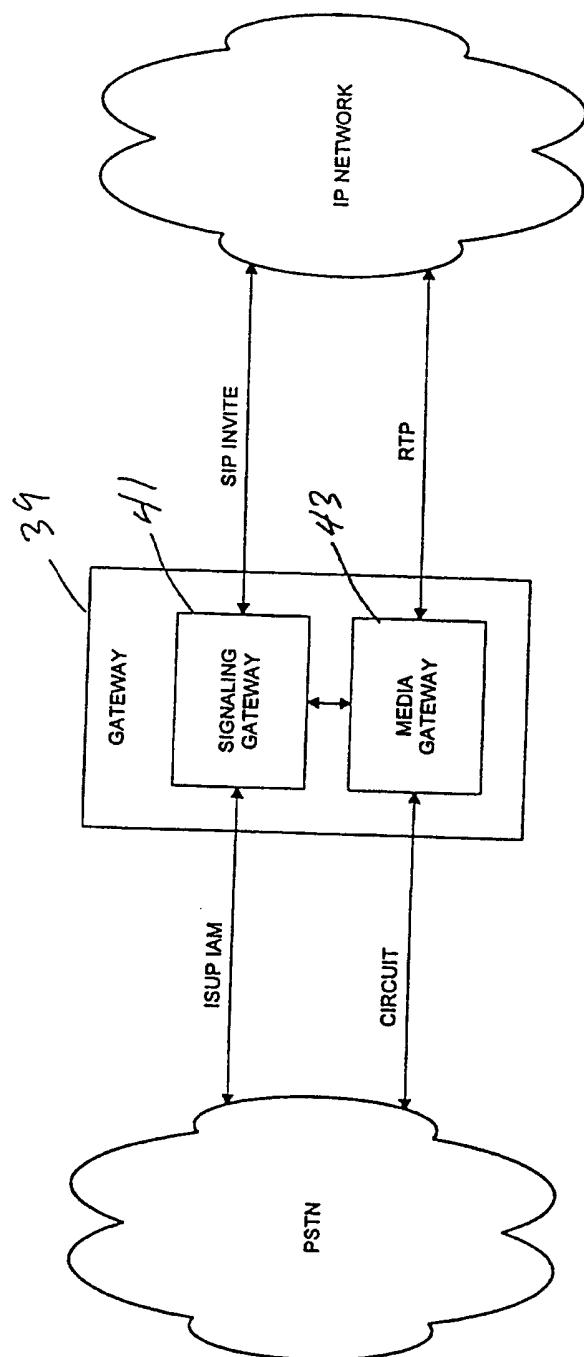5   indicator to a telephony signaling protocol numbering plan indicator.

FIG. 1

FIG. 2

A. CLASSIFICATION OF SUBJECT MATTER

IPC(7) :H04Q 3/00
US CL :370/260, 264, 345, 353, 355, 356, 465, 466, 467

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 370/260, 264, 345, 353, 355, 356, 465, 466, 467

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X --- Y	Schulzrinne et al. Signaling for Internet Telephony, IEEE, September 1998, see pages 298-307.	1-24
X --- Y	WO 97/22209 A (LOW) 19 JUNE 1997, see Figs. 8-11 and 13 and pages 29-33.	1-24

☐ Further documents are listed in the continuation of Box C.   ☐ See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be of particular relevance		
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"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

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DUONG, FRANK |
| Facsimile No. (703) 305-3230 | Telephone No. (703) 308-5428 |

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